CLAIMS

What is claimed is:

1. An Internet telephony conferencing server system comprising a plurality of conferencing centers, each configured for hosting a conference call amongst a plurality of the telephony clients, and wherein each telephony client is configured to measure at least one quality of service characteristic of communication with each conferencing center and to select one of the conferencing centers based on the quality of service characteristic for hosting a conference call.

2. The Internet telephony conferencing server system of claim 1, wherein each telephony client is configured to exchange both audio and video data with the selected one of the conferencing centers.

3. The Internet telephony conferencing server system of claim 2, wherein the telephony client measures the at least one quality of service characteristic by sending a plurality of ping packets to a conferencing server at each conferencing center, receiving a plurality of ping response packets, and measuring latency time and packet loss for each conferencing server.

4. The Internet telephony conferencing server system of claim 3, wherein the telephony client selects one of the conferencing centers by selecting the conferencing server providing the lowest packet loss and if two or more servers have the lowest packet loss, selecting the one of such two or more servers which has the lowest latency time.

5. The Internet telephony conferencing server system of claim 1, wherein the telephony client measures the at least one quality of service characteristic by sending a plurality of ping packets to a conferencing server at each conferencing center,

receiving a plurality of ping response packets, and measuring latency time and packet loss for each conferencing server.

6. The Internet telephony conferencing server system of claim 5, wherein the telephony client selects one of the conferencing centers by selecting the conferencing server providing the lowest packet loss and if two or more servers have the lowest packet loss, selecting the one of such two or more servers which has the lowest latency time.

7. An Internet telephony client for participating in Internet telephony conference calls hosted by a conferencing bridge, the client comprising:

a) an audio interface for:

i) receiving microphone input of an operator speaking and generating digital audio data; and

ii) receiving digital audio data representing a remote voice stream and generating an analog audio signal driving a speaker;

b) an Internet telephony application for:

i) measuring at least one quality of service characteristic of each of a plurality of conferencing servers and selecting the one of the conferencing servers which provides the highest quality of service characteristic for hosting a conference call;

ii) compressing the digital audio data into a sequence of UDP datagrams for sending to the selected conferencing server; and

iii) decompressing a sequence of UDP datagrams from the selected conferencing server to generate the digital audio data representing the remote voice stream.

8. The Internet telephony client of claim 7, further including a video interface for receiving video camera input and for receiving digital video data representing remote video camera input via the Internet telephony application and generating a video

signal for driving a monitor, and the Internet telephony application further operates for compressing the video camera input into a sequence of UDP datagrams for sending to the selected conferencing server and decompressing a sequence of UDP datagrams for the selected conferencing server to generate the video signal.

9. The Internet telephony client of claim 8, wherein the application measures the quality of service characteristic by sending a plurality of ping packets to each conferencing server, receiving a plurality of ping response packets, and measuring latency time and packet loss for each conferencing server.

10. The Internet telephony client of claim 9, wherein the application selects one of the conferencing servers by selecting the conferencing server providing the lowest packet loss and if two or more servers have equivalent packet loss, selecting such two or more servers which has the lower latency time.

11. The Internet telephony client of claim 7, wherein the application measures the quality of service characteristic by sending a plurality of ping packets to each conferencing server, receiving a plurality of ping response packets, and measuring latency time and packet loss for each conferencing server.

12. The Internet telephony client of claim 11, wherein the application selects one of the conferencing servers by selecting the conferencing server providing the lowest packet loss and if two or more servers have equivalent packet loss, selecting such two or more servers which has the lower latency time.

13. The Internet telephony client of claim 12, wherein the UDP datagrams representing the digital audio data are sent to the selected conferencing server and the UDP datagrams representing the remote voice stream are received from the selected conferencing server during an Internet telephony media session and the application sets up the media session by at least one of sending and receiving a

signaling request and by negotiating UDP channels for sending and receiving UDP datagrams.

14. The Internet telephony client of claim 13, wherein the signaling request is a Q.931 signaling request and wherein the UDP channel negotiating is compliant with an H.245 protocol.

15. A method of initiating an Internet telephony conference call, the method comprising:

a) measuring at least one quality of service characteristic of each of a plurality of conferencing servers; and

b) initiating the Internet telephony conference call to one of the conferencing servers based on the quality of service characteristic for hosting a conference call.

16. The method of initiating an Internet telephony conference call of claim 15, wherein the step of initiating the Internet telephony conference call includes selecting the conference server providing the best quality of service by sending a plurality of ping packets to each conferencing server, receiving a plurality of ping response packets, and measuring latency time and packet loss for each conferencing server.

17. The method of initiating an Internet telephony conference call of claim 16, wherein step of selecting the conferencing server providing the best quality of service includes selecting the conference server with the lowest packet loss and if two or more servers have the lowest packet loss, selecting one of such two or more servers which has the lower latency time.

18. The method of initiating an Internet telephony conference call of claim 17, wherein the step of initiating the conference call further includes sending a list of call

participants to the conference server such that a conference bridge associated with the conference server can establish an Internet telephony session with each server.

19. The method of initiating an Internet telephony conference call of claim 18, wherein establishing an Internet telephony session includes opening a Q.931 signaling connection.

20. The method of initiating an Internet telephony conference call of claim 19, wherein establishing an Internet telephony session further includes opening an H.245 session.

21. The method of initiating an Internet telephony conference call of claim 20, wherein establishing an Internet telephony session further includes negotiating UDP datagrams channels for sending and receiving UDP datagrams representing conference audio streams.